REMARKS

In view of the above amendments and the following remarks, reconsideration of the outstanding office action is respectfully requested.

Claim 1 has been canceled without prejudice. Upon entry of the above amendments, Claims 2 to 9 are pending in the application. No new matter has been added by the amendments.

The rejection of Claim 1 under 35 U.S.C.§102(b) as anticipated by U.S. Patent No. 5,742,734 to DeJaco et al. ("DeJaco") is respectfully traversed in view of the cancellation of the claim.

The rejection of Claims 4 and 6-9 under 35 U.S.C.§103(a) for obviousness over DeJaco in view of U.S. Patent No. 6,029,126 to Malvar ("Malvar") is respectfully traversed.

Certain types of sounds, such as sounds of a percussion instrument, include spectrum components that tend to be perceived as noises, which are encoded in a low bit rate, by audio codecs; music including this type of sound is frequently paused. Audio codecs also consider sounds having low amplitudes as noises, which also degrade the sound quality. In order to prevent the audio data from being encoded in a low bit rate, the present invention decides an interval of the audio data to be encoded in a low bit rate by the codec, regardless of whether the audio data are noises or not, and then adjusts amplitude of the audio data in the interval, such that the audio data in the interval are encoded in a bit rate higher than or equal to the low bit rate.

DeJaco discloses a method and apparatus for determining an encoding rate in a variable rate vocoder. In DeJaco, the encoding rate of an input signal is selected based on whether a speech is present in the input signal (col. 2, lines 10-42). In order to determine the presence of speech in the input signal, the input signal in

DeJaco is first filtered into a low frequency component and a high frequency component and the filtered components of the input signal are then individually analyzed to detect the presence of the speech. In DeJaco, the reason for filtering the input signal is to distinguish unvoiced sounds having a high frequency component, which may be determined as a noise, from the background noise.

However, DeJaco does not in any way teach or suggest adjusting the amplitude of audio data of the decided interval before the audio data is processed by the codec, such that the audio data in the interval may be encoded in a bit rate higher than or equal to said low bit rate when processed by the codec, as set forth in Claims 4 and 6 of the present application. More specifically, DeJaco does not teach or suggest a "method for preprocessing audio data to be processed by a codec having variable coding rate" including the step of adjusting the amplitude of audio data of the decided interval before the audio data is processed by the codec, such that the audio data in the interval may be encoded in a bit rate higher than or equal to said low bit rate when processed by the codec, as required by Claim 4 of the present application. Nor does DeJaco teach or suggest an "apparatus for providing audio data encoded by a codec having variable encoding rate" including the means for adjusting the amplitude of audio data of the decided interval before the audio data is processed by the codec, such that the audio data in the interval may be encoded in a bit rate higher than or equal to said low bit rate when processed by the codec, as required by Claim 6 of the present application.

Malvar discloses a system and method for enabling scalable encoding and decoding of digitized audio signals. Malvar discloses that audio paths used with current codecs may include, prior to processing by the codecs, a signal enhancement module, such as automatic gain control , noise reducers, etc (col. 2, lines 41-56). As it is well known, automatic gain control in hands-free teleconferencing is used to automatically adjust the speech level of an audio signal to a predetermined value. In hands-free teleconferencing, for example, audio signals come from different

locations, each signal having its own level. Thus, the automatic gain control can be used to maintain the speech levels from these various sources at a common level so that subsequent processing operates on signals within a specified dynamic range.

However, Malvar does not in any way teach or suggest adjusting the amplitude of audio data of the decided interval before the audio data is processed by the codec, such that the audio data in the interval may be encoded in a bit rate higher than or equal to said low bit rate, as set forth in Claims 4 and 6 of the present application. More specifically, Malvar does not teach or suggest a method for preprocessing audio data to be processed by a codec having variable coding rate including the step of adjusting the amplitude of audio data of the decided interval before the audio data is processed by the codec, such that the audio data in the interval may be encoded in a bit rate higher than or equal to said low bit rate when processed by the codec, as required by Claim 4 of the present application. Nor does Malvar teach or suggest an apparatus for providing audio data encoded by a codec having variable encoding rate, including the means for adjusting the amplitude of audio data of the decided interval before the audio data is processed by the codec, such that the audio data in the interval may be encoded in a bit rate higher than or equal to said low bit rate when processed by the codec, as required by Claim 6 of the present application.

There is no basis for combining the teachings of DeJaco and Malvar. In Malvar, the automatic gain control is merely used to maintain the various speech levels at a common level. This is far different from filtering the input signal into a low frequency component and a high frequency component and then individually analyzing the filtered components of the input signal to detect the presence of the speech. Accordingly, Malvar is not properly combinable with DeJaco. In other words, maintaining the various speech levels at a common level and filtering the input signal into different components and then individually analyzing the filtered components of the input signal are so different from one another that one of

ordinary skill in the art would have no reason to combine teachings from one of these areas and apply them to the other.

Regarding Claim 7, the claimed method involves adjusting an amplitude of a frame in the audio data determined as noise signal, such that the adjusted frame is not determined as noise when processed by the codec. Thus, the frame determined as noise can be encoded in a high bit rate. In contrast, the method of DeJaco involves determining the encoding rate based on whether the speech or music is present in the input signal. The method of DeJaco involves filtering the input signal into a low frequency component and a second frequency component in order to distinguish unvoiced sounds from the background noise. That is, the method of DeJaco discloses distinguishing the unvoiced sounds, which may be encoded at a lower bit rate, from the background noise, whereas the present invention according to Claim 7 discloses adjusting a frame corresponding to noise such that the frame is not determined as noise. Thus, DeJaco does not in any way teach or suggest <u>adjusting an amplitude of the frame corresponding to noise such that the adjusted frame is not determined as noise</u>, as set forth in Claim 7 of the present application. More specifically, DeJaco does not teach or suggest a method for preprocessing audio data to be processed by a codec having variable coding rate including the step of <u>adjusting the amplitude of a frame determined as noise signal such that the adjusted frame is not determined as noise</u> when processed by the codec, as required by Claim 7 of the present application.

Further, Malvar is distinguishable from the present invention in that it teaches using automatic gain control to maintain the various speech levels at a common level and does not in any way teach or suggest <u>adjusting an amplitude of the frame corresponding to noise such that the adjusted frame is not determined as noise</u>, as set forth in Claim 7 of the present application. More specifically, Malvar does not teach or suggest a method for preprocessing audio data to be processed by a codec having variable coding rate including the step of <u>adjusting the amplitude of</u>

a frame determined as noise signal such that the adjusted frame is not determined as noise when processed by the codec, as required by Claim 7 of the present application. For reasons already noted above, Malvar is not properly combinable with DeJaco.

Regarding Claim 8, the claimed method involves adjusting an amplitude of audio data before the audio data is transmitted through the transmission channel, such that the audio data is processed in the codec in a higher bit rate from the bit rate without the adjusting. While it is well known that a codec compresses audio data before it is transmitted through a transmission channel, the compression of audio signal by the codec disclosed in DeJaco is different from adjusting the amplitude of an audio data. DeJaco filters the input signal into a low frequency component and a high frequency component in order to detect presence of a speech including unvoiced sounds, which may be encoded at a lower bit rate in the previous speech coding systems. Thus, DeJaco does not in any way teach or suggest adjusting an amplitude of audio data before the audio data is transmitted through the transmission channel, such that the audio data is processed in the codec in a higher bit rate from the bit rate without the adjusting, as set forth in Claim 8 of the present application. More specifically, DeJaco does not teach or suggest a method for preprocessing audio data to be transmitted through a transmission channel and then to be processed by a codec having variable coding rate including the step of adjusting an amplitude of audio data before the audio data is transmitted through the transmission channel, such that the audio data is processed in the codec in a higher bit rate from the bit rate without the adjusting, as required by Claim 8 of the present application.

Further, Malvar is distinguishable from the present invention in that it teaches using automatic gain control to maintain the various speech levels at a common level and does not teach or suggest adjusting an amplitude of audio data before the audio data is transmitted through the transmission channel, such that the

audio data is processed in the codec in a higher bit rate from the bit rate without the adjusting, as set forth in Claim 8 of the present application. More specifically, Malvar does not teach or suggest a method for preprocessing audio data to be transmitted through a transmission channel and then to be processed by a codec having variable coding rate including the step of <u>adjusting an amplitude of audio data before the audio data is transmitted through the transmission channel, such that the audio data is processed in the codec in a higher bit rate from the bit rate without the adjusting</u>, as required by Claim 8 of the present application. For reasons already noted above, Malvar is not properly combinable with DeJaco.

Regarding Claim 9, the claimed apparatus includes means for adjusting the amplitude of the audio data, such that the audio data is processed in the codec in higher bit rate from the bit rate without the amplitude adjustment. In contrast, DeJaco filters the input signal into a low frequency component and a high frequency component in order to detect the presence of a speech including unvoiced sounds, which may be encoded at a lower bit rate in the previous speech coding systems. Thus, DeJaco does not in any way teach or suggest <u>adjusting the amplitude of the audio data, such that the audio data is processed in the codec in higher bit rate from the bit rate without the amplitude adjustment</u>, as set forth in Claim 9 of the present application. More specifically, DeJaco does not teach or suggest an apparatus for preprocessing audio data to be processed by a codec having variable coding rate, the apparatus being apart from the predetermined codec, including means for <u>adjusting the amplitude of the audio data, such that the audio data is processed in the codec in higher bit rate from the bit rate without the amplitude adjustment</u>, as required by Claim 9 of the present application.

Further, Malvar is distinguishable from the present invention in that it teaches using automatic gain control to maintain the various speech levels at a common level and does not teach or suggest adjusting the amplitude of the audio data, such that the audio data is processed in the codec in higher bit rate from the bit

rate without the amplitude adjustment, as set forth in Claim 9 of the present application. More specifically, Malvar does not teach or suggest an apparatus for preprocessing audio data to be processed by a codec having variable coding rate, the apparatus being apart from the predetermined codec, including means for adjusting the amplitude of the audio data, such that the audio data is processed in the codec in higher bit rate from the bit rate without the amplitude adjustment, as required by Claim 9 of the present application. For reasons already noted above, Malvar is not properly combinable with DeJaco.

For all of these reasons, the rejection of Claims 4 and 6-9 based on the combination of DeJaco and Malvar is improper and should be withdrawn.

The rejection of Claims 2 and 3 under 35 U.S.C.§103(a) for obviousness over DeJaco in view of Malvar and further in view of U.S. Patent No. 4,539,526 to Davis ("Davis") is respectfully traversed.

The methods of the claimed invention involve classifying the audio data to determine whether or not to perform preprocessing of all frames or selected frames of the audio data before the audio data are subject to the codec. In addition, the present invention performs the AGC preprocessing of frames such that the AGC preprocessed frames are encoded in a higher bit rate compared to the bit rate without the preprocessing. In contrast, as described above, the filtering of the input signal in DeJaco is to determine whether the speech including the unvoiced sounds is present in the input signal, Thus, DeJaco does not in any way teach or suggest, in case the audio data includes monophonic sound, performing AGC (automatic gain control) preprocessing of all frames before the audio data is subject to the codec, such that all frames are processed in the codec in a higher bit rate from the bit rate without the preprocessing and, in case the audio data includes polyphonic sound, performing AGC preprocessing of selected frames before the audio data is subject to the codec, such that the selected frames are processed in the

codec in a higher bit rate from the bit rate without the preprocessing, as set forth in Claim 2 and 3 of the present application. More specifically, DeJaco does not teach or suggest a method for preprocessing audio data to be processed by a predetermined codec having variable coding rate including the steps of, in case the audio data includes monophonic sound, performing AGC (automatic gain control) preprocessing of all frames before the audio data is subject to the codec, such that all frames are processed in the codec in a higher bit rate from the bit rate without the preprocessing and, in case the audio data includes polyphonic sound, performing AGC preprocessing of selected frames before the audio data is subject to the codec, such that the selected frames are processed in the codec in a higher bit rate from the bit rate without the preprocessing, as required by Claims 2 and 3 of the present application.

Further, Malvar does not teach or suggest performing AGC preprocessing of all frames or selected frames of the audio data before the audio data are subject to the codec, such that all frames or the selected frames are proeessed in the codec in a higher bit rate from the bit rate without the preprocessing, as set forth in Claims 2 and 3 of the present application. More specifically, Malvar does not teach or suggest a method for preprocessing audio data to be processed by a predetermined codec having variable coding rate including the steps of, in case the audio data includes monophonic sound, performing AGC (automatic gain control) preprocessing of all frames before the audio data is subject to the codec, such that all frames are processed in the codec in a higher bit rate from the bit rate without the preprocessing and, in case the audio data includes polyphonic sound, performing AGC preprocessing of selected frames before the audio data is subject to the codec, such that the selected frames are processed in the codec in a higher bit rate from the bit rate without the preprocessing, as required by Claims 2 and 3 of the present application.

Davis is cited as disclosing a system that performs preemphasis on a signal prior to encoding or decoding, where preemphasis is based on a ratio of high frequency energy to low frequency energy. However, Davis does not teach or suggest performing the AGC preprocessing of frames such that the AGC preprocessed frames are encoded in a higher bit rate rather than reducing noise from the frames, as set forth in Claims 2 and 3 of the present application.

There is no basis for combining the teachings of DeJaco, Malvar, and Davis. In Malvar, the automatic gain control is merely used to maintain the various speech levels at a common level. In Davis, preemphasis is used to amplify the magnitude of select frequency components of an electrical signal for reducing noise This is far different from filtering the input signal into a low frequency component and a high frequency component and then individually analyzing the filtered components of the input signal to detect the presence of the speech. Accordingly, DeJaco, Malvar, and Davis are not combinable with each other. In other words, maintaining the various speech levels at a common level or amplifying the magnitude of select frequency components of an electrical signal for reducing noise and filtering the input signal into different components and then individually analyzing the filtered components of the input signal are so different from one another that one of ordinary skill in the art would have no reason to combine teachings from Malvar and/or Davis and apply them to DeJaco.

Therefore, the rejection of Claim 2 and Claim 3 (which is dependent on Claim 2) based on the combination of the above references is improper and should be withdrawn.

The rejection of Claim 5 under 35 U.S.C.§103(a) for obviousness over DeJaco in view of Malvar as applied to Claim 4 above and further in view of U.S. Patent No. 4,912,766 to Forse ("Forse") is respectfully traversed.

Forse is cited for teaching a system that uses automatic gain control in a speech application, where the system inputs a speech signal, determines spectral parameters, stores gain coefficients for each spectral parameter and then uses the lowest of the gain coefficients to adjust the magnitude of the spectral parameters. However, even if this is true, Forse does not overcome the above-noted deficiencies of DeJaco and Malvar.

Thus, the rejection of Claim 5 (which is dependent from Claim 4) for obviousness over DeJaco in view of Malvar and Forse is improper and should be withdrawn.

In view of all of the foregoing, applicant submits that this case is in condition for allowance and such allowance is earnestly solicited.

CONCLUSION

Applicant respectfully requests that a timely Notice of Allowance be issued in this case.

If any fees, including extension of time fees or additional claims fees, are due as a result of this response, please charge Deposit Account No. 19-0513. This authorization is intended to act as a constructive petition for an extension of time, should an extension of time be needed as a result of this response. The examiner is invited to telephone the undersigned if this would in any way advance the prosecution of this case.

Respectfully submitted,

Date: December 3, 2007

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